MMT Workshop

July 1st, 2009
Queen Mary University of London, London, UK

14:00 ~ 15:30 Session I. Industry Experiences

(moderator : Young-Kwon Lim, net&tv Inc)

Use of MPEG-2 Transport in Broadcast and other applications – Challenges to be met by MMT

Sam Narasimhan (Motorola)

This short presentation provides an overview of MPEG-2 transport standard which was developed over 15 years ago and its (continued) use in majority of Broadcast, DVD and IP based applications. In addition to providing a robust transport mechanism for carriage of various codec's developed by MPEG and other standards bodies, MPEG-2 transport is also used as a foundation for specifications related to physical layer of networks (FEC) and for conditional access (CA). Cable modem standards use MPEG-2 TS for transmission of IP data while DVD specifications use the program stream part of MPEG-2 systems for content coding. With the explicit mechanisms for audio/video synchronization in MPEG-2 TS, majority of the IPTV applications continue to use MPEG-2 TS as the underlying transport layer below IP protocols. The presentation will cover some of these application examples. The presentation will list a set of requirements for MMT so that it can provide the functionalities of MPEG-2 TS (that may still be required in future) and include additional functionalities to overcome some issues we are currently seeing with MPEG-2 transport (that need to be addressed in a new standard).

MMT considering service environments

Jeayeon Song (Samsung)

For specifying the requirements and position in the market of MMT, the service environments and trends are analyzed in this presentation. Streaming service environments have been changed from only legacy broadcasting to peer-to-peer UCC streaming. There are several transport protocols in the each network such as mobile, broadcasting and furthermore, the convergence of service and device already started, the cross-domain will be considered. And according to the user demand of the premium contents providing rich UX, the service trend can affect on MMT structure will be presented such as the post HD, rich interactivity and metadata, etc.

On MPEG Media Transport

Dave Singer (Apple)

This paper will discuss about following issues regarding MPEG media transport

- Use Cases and environments
- Critique
- Needs

DVB experiences and related standards on using MPEG transport mechanisms

Alexander Adolf and Thomas Stockhammer (DVB)

This presentation will introduce various experiences of defining and using application standards based on MPEG transport mechanism including

- Download and random access of MP4 files
- MPEG TS Transport between heterogeneous network
- Cross-layer designs to improve the Quality of Service/Experience (QoS/QoE)
- Context- and Content-Aware Networks
- Internet TV Content Delivery

15:30 ~ 16:00 Coffee Break

16:00 ~ 17:30 Session II. Challenges

(moderator : Jörn Ostermann, University of Hannover)

Fully Interoperable Streaming of Media Resources in Heterogeneous Environments

Michael Eberhard, Christian Timmerer, and Hermann Hellwagner

This paper presents an interoperable multimedia delivery framework for (scalable) media resources based on various MPEG standards and IETF Requests for Comments (RFC). It can be used to transmit (scalable) media resources within heterogeneous usage environments where the properties of the usage environment (e.g., terminal/network capabilities) may change dynamically during the streaming session. The usage environment properties are signaled by interoperable description formats provided by the MPEG-21 Digital Item Adaptation (DIA) standard and encapsulated within the MPEG Extensible Middleware's (MXM) request content protocol. Furthermore, the available media resources are queried by means of the MPEG Query Format (MPQF). Additionally, the actual adaptation and delivery of the content is done by exploiting a state-of-the-art multimedia framework such as that provided by VideoLAN Client (VLC).
http://www-itec.uni-klu.ac.at/~m1eberha/demo

Media-Aware Network Elements on Legacy Devices

Ingo Kofler, Robert Kuschnig, and Hermann Hellwagner

Recent advances in video coding technology like the scalable extension of the MPEG-4 AVC/H.264 video coding standard (H.264/SVC) pave the way for computationally cheap adaptation of video content. In the course of our research we developed a lightweight RTSP/RTP proxy that enables in-network stream processing. Based on an off-the-shelf wireless router (Linksys WRT 54 GL Broadband Router) that runs a Linux-based firmware we demonstrate that the video adaptation can be performed on-the-fly directly on a network device. By utilizing the RTP packetization of the video stream the proxy can adapt the video in the spatial, temporal and SNR domains. The proxy was developed from scratch in ANSI C and was deployed on the router by using the popular openWrt distribution.
http://www-itec.uni-klu.ac.at/~inkofler/demo/

Harmonization with the current QoS protocols for MMT

Doug Young Suh, Jin Woo Hong

This presentation describes how MMT will harmonize the MPEG tools with the QoS protocols of IETF, 3GPP, and IEEE802 series. Such harmonization will enable to exploit various useful tools developed by the related standard development organizations.

Predictable Loss and Predictable Delay for IP media services

Prof. Dr.-Ing. Thorsten Herfet; M.Sc. Manuel Gorius

Internet Protocol based infrastructures become increasingly important for the distribution of digital broadcast media. Unfortunately, available transport protocols do not meet the requirements of such media either concerning the timeliness, the reliability, or the transmission overhead. Of course, HTTP over TCP is currently the prevalent configuration for audiovisual streaming in the Internet as it provides a convenient solution with end-to-end reliability and NAT traversal. However, the protocol is neither suitable for real-time transmission due to its flow control nor does it provide the scalability for large broadcast scenarios. Therefore, current IPTV services as well as IP based mobile broadcast solutions such as 3GPP streaming are based on UDP, usually extended by RTP. Even though multicast is still an open issue on the Internet, this protocol combination at least provides the essential scalability. Nevertheless, as soon as it comes to wireless transmission (802.11, WiMAX, 3GPP), the lack of reliability seriously affects the rendering quality at the receiver since the services suffer from packet loss rates of several percent.
We chose an Adaptive Hybrid Error Correction (AHEC) approach as a basis for our media oriented transport architecture. This highly flexible composition of NACK based ARQ and adaptive packet-level FEC leads to near-optimal coding efficiency as it is controlled by analytical parameter derivation based on a statistical channel prediction model. The ability to fit to certain delay and reliability constraints even allows the parameter optimization beyond the end-to-end connection granularity: Wired and wireless networks usually significantly differ in terms of packet loss. On the other hand, home network segments provide a much lower round trip delay than IP based delivery networks. Obviously, pure end-to-end error correction schemes are not efficient in such heterogeneous network environments. Therefore, our AHEC scheme offers a link-level operation mode which relieves reliable links from the redundancy required for more unreliable links.
http://www.nt.uni-saarland.de/publications